## CLAIMS

We claim:

1. A system architecture for an internet telephone gateway server, comprising:

   hardware for interfacing with the internet and a public switched telephone network; and

   software for connecting telephone calls between transmitters and receivers, said software having the capability of dynamically changing a level of redundancy of a forward error correction algorithm from packet-to-packet in a data stream so as to accommodate data dropouts,

   whereby aural data in a packet is entirely duplicated to maintain the voice quality present prior to the data dropout.

2. The system architecture of claim 1, wherein said gateway server supports full duplex voice transmission with a latency of less than 500 milliseconds.

3. The system architecture of claim 1, wherein said software has the capability of dynamically varying the size or bundling of a data packet from packet-to-packet.

4. The system architecture of claim 1, wherein said software has the capability of dynamically varying from one codec to another codec from packet-to-packet.

5. The system architecture of claim 1, wherein said software varies the size or bundling of data packets from packet-to-packet.

6. A method for eliminating packet losses in an Internet telephone communication, comprising the steps of:

receiving an Internet telephone call from a personal computer or a telephone;

digitizing said telephone call into a digital data stream;

determining a level of redundancy of a forward error correction algorithm based upon Internet conditions, the level of redundancy varying between $k = 0$ and M; and

applying said level of redundancy to a part of said digital data stream,

wherein an $i^{th}$ data packet is repeated $k$ times in said digital data stream, and

whereby aural data in the $i^{th}$ data packet is duplicated to maintain the voice quality present prior to the packet loss.